Digital Audio Broadcasting

Performance of the iBiquity Digital FM IBOC System in Unimpaired Channel Conditions

Test Plan and Procedures

Document No. 02-07B

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1 Introduction

1.1 General Overview

1.1.1 Background

Over the course of the past year, the Advanced Television Technology Center (ATTC) has conducted an extensive laboratory test program designed to fulfill the requirements of the NRSC (National Radio Systems Committee) AM and FM IBOC DAB system evaluation and standardization processes.

This test program was conducted using the "first generation" of IBOC transmitters and receivers produced by iBiquity Digital. These first generation units incorporated an audio coding algorithm commonly referred to as MPEG AAC. However, iBiquity has announced that in the *final* IBOC systems, the MPEG AAC compression algorithm will be replaced with an iBiquity proprietary compression algorithm referred to as PAC (Perceptual Audio Coding). Consequently, certain elements of the NRSC test program were postponed until the PAC compression algorithm could be incorporated in the next generation of iBiquity IBOC transmitters and receivers.

Recently, iBiquity provided ATTC with a transmitter and receiver that incorporated the new PAC compression algorithm into the FM IBOC system. In this portion of the test program, ATTC will use this hardware to test the audio performance of the PAC compression algorithm under *unimpaired* RF channel conditions. These tests shall be conducted in accordance with Section I of the NRSC FM laboratory test procedures documentation.

1.1.2 Document Scope

This document contains all necessary test methodologies and procedures to be followed during execution of this test series. In addition to methodology descriptions, details regarding equipment setup and test execution are specified.

1.1.3 Related Documents

A summary of test results may be found in:

ATTC Doc. 02-08, Digital Audio Broadcasting, Performance of the iBiquity Digital FM IBOC System in Unimpaired Channel Conditions, Summary of Test Results, February 2002.

Additional details regarding the construction and performance of the ATTC IBOC DAB test platform may be found in:

Digital Audio Broadcasting, Test Bed Proof of Performance Record, Document No. 01-01, Revision 1.0, January 2001, Advanced Television Technology Center, Inc.

Digital Audio Broadcasting, Test Bed Proof of Performance Plan, Document No. 01-20, Revision 2.0, November 2001, Advanced Television Technology Center, Inc.

1.2 Methodology Overview

1.2.1 Objectives

The test procedures in this document have one primary objective, which may be summarized as:

1. Generate digital audio recordings that represent the audio performance of iBiquity's 2nd generation transmitter and receiver technology, incorporating PAC technology, in unimpaired channel conditions.

1.2.2 Channel Conditions

The scope of this test effort is limited to unimpaired RF channel conditions. This condition is chosen so that the performance of the PAC compression technology may be evaluated in error-free conditions. Although the unimpaired RF channel condition does not likely represent the "real-world", it does allow the audio quality of the system to be evaluated in an isolated manner, without the influence of additional, unnecessary test variables.

1.2.3 Evaluation Methods

It is expected that subjective evaluation methods will be used to score each digital audio recording. The subjective evaluation listening tests will be performed at Dynastat, Inc., and are outside the scope of this test plan.

2 Signal Descriptions

The sections that follow define the signals that are to be used in these tests. Later in this document, if a Desired Hybrid FM signal is required, the definition of such a signal may be found in this section.

2.1 RF Signals

2.1.1 Desired Analog FM

In all cases, a desired analog FM signal shall have the following characteristics:

1) Main channel modulation:
   a) Stereo transmission
   b) 75μs pre-emphasis
   c) 10% pilot injection
   d) Test dependent audio
      i) Appropriate critical listening material peaking at 90% modulation (67.5kHz deviation) with dynamic range processing consistent with the musical genre of the audio. Pilot contributes 10% for total modulation = 100% (Refer to Appendices B & C for the audio cut list and corresponding processor presets).
2) Subcarriers: None
3) Main Carrier:
   a) 97.9 MHz
4) Power
   a) Strong: -47dBm

2.1.2 Desired Hybrid FM

A Desired Hybrid FM signal is defined as the spectral sum of an analog desired signal (as described in 2.1.1) and the digital carriers as generated by an iBiquity Digital 2nd Generation IBOC exciter in hybrid mode. The digital carriers utilize OFDM modulation, and the audio undergoes rate reduction/compression as implemented by the iBiquity Digital PAC algorithm. The sum of *all* digital carriers in the hybrid signal shall have an *average* power that is 20 dB less than the average analog power.

3 Standard Methodologies

3.1 FM Band RF Measurements

3.1.1 Analog FM Power

Methodology

This procedure specifies the method for measuring analog FM power. An FM signal has the rather unique characteristic of constant power regardless of the content or amplitude of the modulating signal. Therefore, power can be measured with or without a modulating signal present.

If a modulating signal *is* present, then the power across the entire channel is integrated in order to determine overall power. Traditionally, there are three common laboratory methods of performing this integration:

1) Numerically, by utilizing a DSP based Vector Signal Analyzer (VSA) or band power markers on a spectrum analyzer
2) Physically, by detecting heat in a thermal sensor
3) Electrically, by using a diode to rectify the signal and take advantage of a diode's "square-law" region of operation

For our testing purposes, we shall use method (3) in conjunction with an average power meter and RMS responding diode detector.

Setup

The measuring instrument shall be an HP437B average power meter with an 8481D diode detection sensor, and will be configured as shown in Table 3-1.

Table 3-1 HP 437B Setup – Analog FM Power

Parameter	Description
Sensor Type	HP 8481D (diode detector)
Limit Checking	On
Low Limit	-70dBm
High Limit	-20dBm
Cal. Factor	98.5%
Note: "Preset" first, and then set above parameters	

Usage of the HP437B must take into consideration the dynamic range of the diode detector. Under no circumstances shall a measurement be taken outside the sensor's measurement range. In addition, extra care must be taken during IBOC measurements due to the high peak-to-average ratio of COFDM. Therefore, measurements with an IBOC signal must maintain 10dB of "headroom" below the sensor's peak range.

In addition, note that measurements must be made under conditions with no interferers present since any out of band signal will artificially increase readings on the power meter.

Procedure

1) Configure the instrument according to the tables found in the "Setup" section above.
2) The analog carrier can be either modulated or unmodulated. There is no procedural change for either case.
3) The power level shall be observed (or collected over the GPIB bus) and the instantaneous reading recorded.

Presentation of Data

The resulting measurement shall be expressed in dBm units with a precision of 0.01dBm.

3.1.2 IBOC Hybrid Mode Power

Methodology

This procedure specifies the method for measuring the power of an IBOC signal in hybrid mode. A hybrid signal is the spectral sum of a traditional analog FM signal and COFDM digital carriers at the channel edges.

The true average power of this signal may be determined by integrating over the entire channel, using an average power meter or vector signal analyzer, as discussed in 3.1.1 above. However, the resulting number would not be easily related to traditional analog power measurements. For comparison purposes, we would like to be able to say that a –30dBm hybrid signal has the same amount of *analog* energy as a traditional –30dBm FM signal.

For this reason, all references to the power of a hybrid signal will actually be specifying power in the signal which results from traditional analog FM modulation. In other words, *a - 30dBm hybrid signal will actually have –30dBm of analog FM energy plus the energy resulting from the digital carriers.*

Consequently, in order to exclude the digital energy from our measurements, it is not possible to directly measure hybrid mode power using a traditional average power meter (at least not without an impracticably steep bandpass filter). Therefore, there are three practical measurement methods that can be employed:

1) Remove digital sidebands and measure remaining analog power with a traditional average power meter or Vector Signal Analyzer
2) Use a Vector Signal Analyzer to numerically integrate over analog bandwidth
3) Use spectrum analyzer to measure power of the *unmodulated* analog carrier.

For our purposes, we will utilize the first method in conjunction with an HP437B average power meter. This is facilitated by the fact that the test bed has electromechanical RF switches, which can readily switch the digital carriers in and out of the spectrum. In addition, the removal of the digital sidebands increases measurement accuracy.

Setup

The measuring instrument shall be an HP437B average power meter with an RMS responding sensor, and will be configured as shown in Table 3-2.

Table 3-2 HP 437B Setup – Hybrid Mode Power

Parameter	Description
Sensor Type	HP 8481D (diode detector)
Limit Checking	On
Low Limit	-70dBm
High Limit	-20dBm
Cal. Factor	98.5%
Note: "Preset" first, and then set above parameters	

Usage of the HP437B must take into consideration the dynamic range of the diode detector. Under no circumstances shall a measurement be taken outside the sensor's measurement range. In addition, note that measurements must be made under conditions with no interferers present since any out of band signal will artificially increase readings on the power meter.

Procedure

1) Configure the instrument according to the tables found in the "Setup" section above.
2) The analog carrier can be either modulated or unmodulated. There is no procedural change for either case.
3) The digital carriers shall be removed by opening the appropriate electromechanical RF switch. The remaining energy will be purely analog FM.
4) The power level shall be observed (or collected over the GPIB bus) and the instantaneous reading recorded.

Presentation of Data

The resulting measurement shall be expressed in dBm, and rounded to the nearest hundredth of a decimal place. It should be made clear that this power refers to the analog energy and does not represent the average power level of the entire hybrid signal (as discussed above).

3.2 Audio Recording and Editing

This subsection defines procedures for performing common audio editing operations, such as digital recording, editing and leveling.

3.2.1 Audio Recording

Methodology

All audio recordings must be made in such a way that no significant artifacts are introduced by the recording process. This necessitates the exclusive use of a digital audio recording format. Furthermore, this format must be uncompressed and able to sustain a data rate that supports multi-track audio with a resolution of 16 bits and a sampling frequency of 44.1kHz. Additionally, the recordings must be made in a manner that lends itself to archival and duplication.

In order to meet these requirements, the DTRS digital tape recording technology of Tascam shall be used. This format records eight tracks of digital audio on the same cassette shell utilized by the popular Hi-8 video format. Tascam's format has also gained widespread popularity within the professional audio recording community, enabling easy exchange of materials across different facilities.

There is also a requirement to unambiguously associate individual “takes” on any given tape with a specific test number or test setup. In order to accomplish this objective, SMPTE timecode shall be employed extensively. The DA-98 recorder has provisions for a ninth track, containing unique SMPTE timecode. This timecode provides the ability to log the contents of a DTRS digital audio tape with resolution to 33.4 milliseconds.

In order to generate this log of tape contents, identical and synchronous time code shall be routed to the DA-98 recorder and an external computer simultaneously. The external computer shall keep track of the current test setup and take a snapshot of the SMPTE timecode at the start of each test. Microsoft Excel Visual Basic scripts shall then be used to generate a log, which will relate SMPTE timecode locations to specific test conditions.

Setup
The recording device shall be a Tascam DA-98 Digital Multitrack Recorder, operating in conjunction with a Tascam IF-AE8 AES digital audio interface. Complete details of the menu setups for these units may be found in the test bed proof-of-performance documentation.

Procedure
Operation of the Tascam DA-98 recorder is straightforward. However, there are several important points which shall be observed by the test engineer throughout the testing process:

1) The recorder shall always be operated with a 44.1kHz sampling frequency and 16 bits of resolution
2) Whenever possible, the digital inputs of the IF-AE8 AES interface shall be used. Normally, sample rate conversion shall be disabled. However, certain sources containing a high level of jitter may require activation of sample rate conversion.
3) For sources available only in an analog format (e.g. radio outputs), the balanced analog inputs of the DA-98 shall be employed, utilizing the unit’s internal A-to-D converters.

Presentation of Data
1) *All* tapes shall be labeled with a unique identification code consisting of the date and an incrementing index number (e.g.: 07-25-01-03 represents the third tape generated on 07-25-01).
2) *All original* test results tapes shall be archived in ATTC’s tape vault, and indexed in ATTC’s tape library database.
3) A log shall be generated for each recording. This log shall relate SMPTE timecode values and tape identification numbers to test setup conditions.

3.2.2 Audio Editing
Methodology
The “raw” test result tapes generated by the test bed are expected to require significant editing. This editing should eliminate periods of silence and any glitches that may occur while equipment setups are being changed in between tests. The editing should occur in a

manner which introduces *no* additional audio artifacts or impairments. Additionally, the editing should preserve the original tape in its entirety. The recording from each test shall be formatted into an individual computer file, which can then be readily played back on the equipment of the subjective evaluation laboratory.

In order to accomplish these objectives, it is necessary to perform the editing in the digital domain on a digital audio workstation. This workstation must provide facilities for transferring DTRS tapes to computer format .wav files. It must also provide professional audio editing software, and a CD data recorder to export the resultant .wav format files.

<u>Setup</u>

A digital audio workstation shall be used for all editing. This workstation shall consist of:

1) High speed PC workstation
2) Lynx One professional audio card w/ AES and word clock I/O
3) Lucid DA9624 external D/A converter and headphone amplifier
4) Sennheiser HD-600 headphones
5) Horita TR-100 stand-alone time code reader
6) Tascam TDIF interface card
7) Tascam DA-98 or DA-78HR DTRS player/recorder
8) Cool Edit Pro professional audio editing software

<u>Procedure</u>

1) Transfer DTRS digital audio tape to the computer hard drive and save in .wav format (for details, refer to: *DTRS Tape to .wav Transfer Procedures, ATTC Doc. 01-18*)
2) Use SMPTE time code to locate each test within the original .wav file
3) Trim/edit the "heads" and "tails" of each test, such that there is absolutely no silence before or after each test.
4) Save this trimmed file, using a filename that conforms to the following format: *ATTCTest#_RadioAbbreviation.wav*

<u>Presentation of Data</u>

All edited audio files shall be transferred to a recordable CD in .wav format. These audio files shall be archived at ATTC with a log relating the .wav filename, the ATTC test number and the actual test setup conditions.

3.2.3 Audio Leveling & File Renaming

<u>Methodology</u>

Once the test results audio has been recorded, transferred to .wav format and edited, there is an additional requirement to "level" this audio for certain applications. The subjective evaluation laboratory has designed experiments that require all audio samples to have approximately the same perceptual loudness. For example, if an experiment participant is required to listen to a group of audio samples, each one of these samples should have the same perceptual loudness as the other samples in the group.

In order to meet this objective, an additional editing step is required. This additional step is referred to as "leveling". The leveling process utilizes audio editing software to adjust the amplitude of audio recordings. A professional audio editor subjectively evaluates the perceived loudness of each sample within a group, and then adjusts all samples to fall within an acceptable range of perceptual loudness.

In addition to this leveling process, the files must also be renamed in such a way that their filename represents the most important parts of the test setup. Although the ATTC test number uniquely and fully identifies each test, a more descriptive, alphanumeric filename is useful for the subsequent data analysis stages. A convention for this descriptive filename is described in the document entitled: *Naming Convention for Subjective Audio Files, ATTC Doc. 01-03*. A Visual Basic computer program shall be used to convert the original filename (e.g. *5102_Delp_Prince.wav*) into the more descriptive filename (e.g. *5102_B_Delp_NONE_X_NONE_X_AWGN_B-2dB_X_PRINCE_ROCK.wav*). This Visual Basic program shall employ a lookup table to relate the ATTC test number in the original filename to the new descriptive filename, and then perform the rename operation.

Setup

The equipment setup shall be identical to the editing setup described in 3.2.2. It is important to note that the external D/A converter and headphones are identical to the equipment used in the listening stations of the subjective evaluation laboratory. In this manner, the professional audio editor evaluates perceptual loudness using an equipment setup that emulates the setup used by experiment participants.

Procedure

1) Determine which audio samples will belong to the experimental group. (This information will be provided by the experiment designer)
2) Copy all of these audio samples into a common directory. (Note that these samples should all have been recorded, transferred and edited in accordance with the procedures of 3.2.1 and 3.2.2.)
3) Use the Windows Commander software application to listen to each audio sample in the experiment.
4) Make notes on the perceptual loudness of each sample, and estimate the relative loudness of this sample compared to the quietest samples in the group. These estimations should be in dB units of attenuation required.
5) Review estimated attenuation values for any discernible pattern. (This pattern may relate to the audio cut, the radio type, the interference conditions, etc…)
6) Adjust each audio sample according to these estimates or any pattern that may have been identified.
7) Repeat steps three thru six until all audio samples in the group have nearly the same perceptual loudness. (Note that the audio editor should constantly strive to minimize the number of amplitude changes that are made to each audio sample, as excessive amplitude adjustments may add a minute amount of noise to the signal.)
8) Use the appropriate Visual Basic script and lookup table to automatically rename the leveled audio files with their more descriptive filename.

Presentation of Data

All leveled audio files shall be transferred to a recordable CD in .wav format. These audio files shall be archived at ATTC, and copies may be sent to the subjective evaluation laboratory. Note that the filenames shall contain the original ATTC test number, so the logs generated from the editing procedure of 3.2.2 may also be used to identify these leveled audio cuts.

4 Digital Performance Procedures

4.1 FM IBOC Performance in Unimpaired Channel Conditions (NRSC I)

Objectives

Generate digital audio recordings that represent the audio performance of iBiquity's 2nd generation transmitter and receiver technology.

Methodology

This test series shall utilize the test platform to simulate unimpaired RF channel conditions. In this "clean channel" condition, the audio outputs of four analog receivers and the iBiquity 2nd generation IBOC receiver will be recorded. The IBOC receiver recordings will represent the subjective performance of the IBOC system in error-free, ideal reception conditions. The recordings of the analog receivers will represent the subjective performance of the conventional analog FM system in ideal reception conditions.

Test Conditions

Table 4-1 enumerates the test conditions to be included in this test series. The following information may also be helpful in understanding the construction of this table:

1. Each row of the table represents a single test.
2. The # column indicates the ATTC assigned test number.
3. The *Desired* column indicates the mode and signal strength of the RF signal.
4. The *Analog Proc.* column indicates the audio processing preset applied to the analog portion of the signal (details of the processor setup for each preset may be found in Appendix C)
5. The *DAB Proc.* column indicates the audio processing preset applied to the DAB portion of the signal. (The phrase "Hard Bypass" indicates that there was no audio processing employed for that particular test.)
6. The *Audio Cut* column identifies the specific audio cut under test (refer to Appendix B for a more descriptive list of the audio cuts)
7. The *Receiver* column indicates the receiver under test. (Note that "4 Analog RX" refers to the receivers listed in Appendix A)

Table 4-1 IBOC Quality Test Conditions (NRSC I)

#	Desired	Analog Proc.	DAB Proc.	Audio Cut	Receiver
1001	Hybrid: Moderate	Light	Hard Bypass	Woman	IBOC
1002	Hybrid: Moderate	Light	Hard Bypass	Man	IBOC
1003	Hybrid: Moderate	Light	Hard Bypass	Brokaw	IBOC
1004	Hybrid: Moderate	Light	Hard Bypass	Bach	IBOC
1006	Hybrid: Moderate	Light	Hard Bypass	Enya	IBOC
1008	Hybrid: Moderate	Light	Hard Bypass	Glockenspiel	IBOC
1011	Hybrid: Moderate	Light	Hard Bypass	Saito	IBOC
1012	Hybrid: Moderate	Light	Hard Bypass	Persian	IBOC
1014	Hybrid: Moderate	Light	Hard Bypass	1812	IBOC
1015	Hybrid: Moderate	Light	Hard Bypass	Trumpet	IBOC
1016	Hybrid: Moderate	Light	Hard Bypass	MMW	IBOC

#	Desired	Analog Proc.	DAB Proc.	Audio Cut	Receiver
1017	Hybrid: Moderate	Medium	Hard Bypass	Simon	IBOC
1018	Hybrid: Moderate	Medium	Hard Bypass	Clapton	IBOC
1023	Hybrid: Moderate	Medium	Hard Bypass	Travis	IBOC
1029	Hybrid: Moderate	Hard	Hard Bypass	Grant	IBOC
1034	Hybrid: Moderate	Light	2B Classical	Carmen	IBOC
1038	Hybrid: Moderate	Light	2B Classical	Messiah	IBOC
1044	Hybrid: Moderate	Hard	Contemporary2-5B	EarthWindFire	IBOC
1045	Analog: Strong	Light	None	Woman	4 Analog Rx
1046	Analog: Strong	Light	None	Man	4 Analog Rx
1047	Analog: Strong	Light	None	Brokaw	4 Analog Rx
1048	Analog: Strong	Light	None	Bach	4 Analog Rx
1049	Analog: Strong	Light	None	Carmen	4 Analog Rx
1050	Analog: Strong	Light	None	Enya	4 Analog Rx
1052	Analog: Strong	Light	None	Glockenspiel	4 Analog Rx
1053	Analog: Strong	Light	None	Messiah	4 Analog Rx
1055	Analog: Strong	Light	None	Saito	4 Analog Rx
1056	Analog: Strong	Light	None	Persian	4 Analog Rx
1058	Analog: Strong	Light	None	1812	4 Analog Rx
1059	Analog: Strong	Light	None	Trumpet	4 Analog Rx
1060	Analog: Strong	Light	None	MMW	4 Analog Rx
1061	Analog: Strong	Medium	None	Simon	4 Analog Rx
1062	Analog: Strong	Medium	None	Clapton	4 Analog Rx
1067	Analog: Strong	Medium	None	Travis	4 Analog Rx
1072	Analog: Strong	Hard	None	EarthWindFire	4 Analog Rx
1073	Analog: Strong	Hard	None	Grant	4 Analog Rx

Procedure

1) The IBOC 2nd generation receiver and the four analog receivers will be used for this test series (see Appendix A).
2) Table 4-1 shall be used as a guide for setting up each test. Each row of the test table shall be executed individually.
3) Establish the signal strength, frequency, and mode of the desired signal as indicated in the test grid. For a complete definition of the desired analog and desired hybrid signals, refer to 2.1.1 and 2.1.2, respectively. The procedures for measuring FM power may be found in 3.1.1 and 3.1.2.
4) Set the audio dynamic range processing of the desired channel analog signal to a preset which corresponds to the test grid specification.
5) Begin recording the output of the IBOC or analog receiver to digital audio tape according to 3.2.1.
6) Play the appropriate audio cut on the desired channel, as indicated in the test grid. Loop this audio cut twice. (For cases where the IBOC receiver is under test, and DAB processing is *not* specified as bypass, use the iBiquity Digital pre-processed CD. In all other cases, use the standard ATTC source material CD.)
7) Perform the audio transfer, editing, leveling and renaming process described in 3.2.2 and 3.2.3.

Presentation of Data

The test results shall consist of digital audio recordings. Each audio cut will be recorded twice. However, the first cut will be discarded, as it is intended to allow the processor sufficient settling time. The remaining audio cut shall be considered the final test result.

A Appendix A - Receivers Under Test

Table A-1 IBOC FM Transmitter and Receiver Under Test

Type	Make	Model	Serial #	Software Revision
Transmitter/ Exciter	iBiquity	2nd Generation – with PAC	----006---	1.03
Receiver	iBiquity	2nd Generation – with PAC	----003---	1.03

Table A-2 Analog FM Receivers Under Test

Type	Make	Model	Serial #
OEM Auto	Delphi	09394139	89DDSTM103490265
Aftermarket Auto	Pioneer	KEH-1900	UHHI086599UC
Home Hi-Fi	Technics	SA-EX110P-K	GX9DA84758
Portable	Sony	CFD-S22	S01-0433905-A

B Appendix B - Audio Cut Lis t

Table B-1 lists the audio cuts used throughout the test procedures. In the test procedures, audio cuts are specified by their "nickname". The table below may be used to relate this nickname to the original source material. The table may also be used to determine what type of dynamic range processing should be applied to each individual audio cut. Note that the dynamic range processing is different, depending on whether the material is to be played through an analog FM or DAB system. Information on the processor settings which implement these different types of dynamic range processing may be found in Appendix C.

In cases where DAB processing is required (note that most cases do not require processing for DAB), the audio material shall be "pre-processed" in advance, by iBiquity Digital, using their Orban 6200-DAB audio processor.

Table B-1 Audio Test Material and Associated Processor Settings

Nickname	Artist/ Composer	Album/ Work	Song/ Movement	Type*	DAB Processing	FM Processing
1812	Tchaikovski (Classical Thunder)	1812 Overture	Track 17	CI	None/Bypass	Light
Bach	Bach	Brandenburg Concerto #5, D Major	Allegro	CI	None/Bypass	Light
Basil	Toni Basil	VH1, More of the Big 80's	Mickey	RFV	Contemp1-5B	Hard
Brokaw	Tom Brokaw	The Greatest Generation	---	SMV	None/Bypass	Light
Carmen	Bizet	Carmen	---	CI	2B-Classical	Light
Clapton	Eric Clapton	Best of Eric Clapton	Change the World	RMV	None/Bypass	Medium
Cole	Paula Cole	Harbinger	Happy Home	RFV	Light5B-20k	Medium
Cray	Moulton Labs	Bang&Olufsen Test Sequence	Roberty Cray	BJM	Contemp1-5B	Medium
Crowded	Crowded House	Woodface	Weather With You	RMV	2B-Classical	Medium
CSNY	Crosby, Stills, Nash & Young	Looking Forward	Sanibel	RMV	Light5B-20k	Medium
Debussy	Debussy	String Quartet in G Minor	Anime et tres decide	CI	None/Bypass	Light
Enya	Enya	Shepherd Moons	Angeles	NAI	None/Bypass	Light
EWF or EarthWindFire	Earth, Wind & Fire	Greatest Hits	Lets Groove	RMV	Contemp2-5B	Hard
Fagen	Donald Fagen	The Nightfly	I.G.Y.	RMV	None/Bypass	Medium
Fleetwood	Fleetwod Mac	Tango in the Night	Big Love	RMXV	2B-Classical	Medium
Glock	Glockenspiel	SQAM Disc	---	CS	None/Bypass	Light
Grant	Amy Grant	Heart in Motion	Baby, Baby	RFV	None/Bypass	Hard
Ibert	Jaques Ibert	Summertime Music for Oboe	Entr'acte	CI	None/Bypass	Light
Man	English Man	SQAM Disc	---	SMV	None/Bypass	Light
Messiah	Handel	Messiah	Hallelujah	CC	2B-Classical	Light
MMW	Medewski, Martin and	Shack Man	Hermeto's Daydream	JI	None/Bypass	Light

Nickname	Artist/ Composer	Album/ Work	Song/ Movement	Type*	DAB Processing	FM Processing
	Wood					
Persian	Saba	Persian Classical Music	The Yellow Sparrow	FI	None/Bypass	Light
REO	REO Speedwagon	Hi Fidelity	Keep on Loving You	RMV	None/Bypass	Hard
Saito	Moulton Labs	Critical Listening Excerpts CD	Kyoko Saito	CF	None/Bypass	Light
Santana	Carlos Santana	Supernatural	Smooth	RMV	Contemp1-5B	Hard
Simon	Paul Simon	Rhythm of the Saints	Can't Run But	RPI	None/Bypass	Medium
Stansfield	Lisa Stansfield	Lisa Stansfield	The Real Thing	RFV	None/Bypass	Hard
Stravinski	Stravinski (Bernstein Conducts)	Firebird	Track 5	CI	2B-Classical	Light
Travis	Randy Travis	A Man Ain't Made of Stone	A Heartache in the Works	CMV	None/Bypass	Medium
Trumpet	Trumpet	SQAM Disc	---	CS	None/Bypass	Light
Vega	Suzanne Vega	Nine Objects of Desire	Caramel	RFV	None/Bypass	Medium
Woman	English Woman	SQAM Disc	---	SFV	None/Bypass	Light

<u>*Type Codes</u>

CI = Classical Instrumental
SMV = Speech Male Vocal
RMV = Rock Male Vocal
BJM = Blues/Jazz Male
CS = Critical Sample
CC = Classical Chorus
CF = Classical Female
CMV = Country Male Vocal
RI = Rock Instrumental
SFV = Speech Female Vocal
NAI = New Age Instrumental
RMXV = Rock Mixed Vocals
RFV = Rock Female Vocal
JI = Jazz Instrumental
RPI = Rock/Pop Instrumental
FI = Folk Instrumental

C Appendix C – FM Audio Processor Settings

The tables in this Appendix provide detailed information on the setup of the FM audio processing equipment for several different "presets". These processing presets are applied to audio material depending on the genre of the audio. A complete list of audio cuts, their genre and their corresponding processor presets may be found in Appendix B.

For analog FM, the audio processing hardware consists of a Cutting Edge Omnia 4500. Settings for the Omnia 4500 FM processor were developed by the NRSC DAB Subcommittee - Test Procedures Working Group. These settings were designed to be *representative* of typical, real-world radio station processing.

Table C-1 Analog FM Processor: Light Preset

Processor Name: Cutting Edge Omnia 4500 Preset Name: Light			
Parameter	**Value**	**Parameter**	**Value**
WB-AGC	IN	**LF-LIMITER**	
AGC Drive	(+)6.0	Drive	0.0
Attack	3	Threshold	(+)2.0
Release	0	Attack	3
Make-Up Gain	1	Release	2
Gate Thresh	4	Hold Thresh	4
BASS		**MF-LIMITER**	
Deep Bass	0.0	Drive	0.0
Phat Bass	0.0	Threshold	0.0
		Attack	3
WARMTH	0.0	Release	1
		Hold Thresh	2
X-OVER			
Low Gain	0.0	**PR-LIMITER**	
Mid Gain	0.0	Drive	0.0
Pres Gain	(+)1.0	Threshold	0.0
High Gain	(+)1.5	Attack	3
		Release	2
LF-AGC		Hold Thresh	4
Attack	2		
Release	0	**HF-LIMITER**	
Make-Up Gain	2	Drive	(+)1.0
Gate Thresh	3	Threshold	(-)7.5
RTP Speed	Slow	Attack	3
RTP Level	(-)10	Release	2
		Hold Thresh	1
MF-AGC			
Attack	3	**MIXER**	
Release	0	Low Band	0.0

Processor Name: Cutting Edge Omnia 4500 Preset Name: Light			
Make-Up Gain	2	Mid Band	0.0
Gate Thresh	3	Pres Band	(-)4.0
RTP Speed	Slow	High Band	(-)5.5
RTP Level	(-)10		
		CLIPPER	
PR-AGC		Drive	(+)0.5
Attack	3		
Release	0	**COMP CLIP**	
Make-Up Gain	3	Drive	0.0
Gate Thresh	2		
RTP Speed	Slow		
RTP Level	(-)10		
HF-AGC			
Attack	4		
Release	1		
Make-Up Gain	3		
Gate Thresh	2		
RTP Speed	Slow		
RTP Level	(-)5		

Table C-2 Analog FM Processor: Medium Preset

Processor Name: Cutting Edge Omnia 4500 Preset Name: Medium			
Parameter	**Value**	**Parameter**	**Value**
WB-AGC	IN	**LF-LIMITER**	
AGC Drive	(+)6.0	Drive	(+)1.5
Attack	3	Threshold	(+)1.5
Release	0	Attack	4
Make-Up Gain	1	Release	2
Gate Thresh	4	Hold Thresh	4
BASS		**MF-LIMITER**	
Deep Bass	(+)4.0	Drive	(+)1.5
Phat Bass	(+)2.0	Threshold	0.0
		Attack	3
WARMTH	(+)1.0	Release	1
		Hold Thresh	3
X-OVER			
Low Gain	(+)2.0	**PR-LIMITER**	
Mid Gain	(+)2.0	Drive	(+)1.5
Pres Gain	(+)3.0	Threshold	0.0
High Gain	(+)4.0	Attack	3

Processor Name: Cutting Edge Omnia 4500 Preset Name: Medium			
		Release	2
LF-AGC		Hold Thresh	2
Attack	2		
Release	0	**HF-LIMITER**	
Make-Up Gain	2	Drive	(+)2.0
Gate Thresh	3	Threshold	(-)7.5
RTP Speed	Slow	Attack	3
RTP Level	(-)10	Release	2
		Hold Thresh	1
MF-AGC			
Attack	2	**MIXER**	
Release	2	Low Band	0.0
Make-Up Gain	2	Mid Band	0.0
Gate Thresh	3	Pres Band	(-)4.0
RTP Speed	Slow	High Band	(-)5.0
RTP Level	(-)10		
		CLIPPER	
PR-AGC		Drive	(+)1.0
Attack	2		
Release	2	**COMP CLIP**	
Make-Up Gain	3	Drive	(+)1.0
Gate Thresh	2		
RTP Speed	Slow		
RTP Level	(-)10		
HF-AGC			
Attack	3		
Release	2		
Make-Up Gain	3		
Gate Thresh	2		
RTP Speed	Slow		
RTP Level	(-)5		

Table C-3 Analog FM Processor: Hard Preset

Processor Name: Cutting Edge Omnia 4500 Preset Name: Hard			
Parameter	**Value**	**Parameter**	**Value**
WB-AGC	IN	**LF-LIMITER**	
AGC Drive	(+)6.0	Drive	(+)2.5
Attack	3	Threshold	(+)1.0
Release	0	Attack	4
Make-Up Gain	1	Release	2
Gate Thresh	4	Hold Thresh	4
BASS		**MF-LIMITER**	

Processor Name: Cutting Edge Omnia 4500 Preset Name: Hard			
Deep Bass	(+)6.0	Drive	(+)2.5
Phat Bass	(+)3.0	Threshold	0.0
		Attack	4
WARMTH	(+)1.0	Release	3
		Hold Thresh	3
X-OVER			
Low Gain	(+)3.5	**PR-LIMITER**	
Mid Gain	(+)3.5	Drive	(+)2.5
Pres Gain	(+)3.5	Threshold	0.0
High Gain	(+)4.0	Attack	4
		Release	3
LF-AGC		Hold Thresh	2
Attack	2		
Release	0	**HF-LIMITER**	
Make-Up Gain	2	Drive	(+)3.0
Gate Thresh	3	Threshold	(-)7.5
RTP Speed	Slow	Attack	3
RTP Level	(-)10	Release	3
		Hold Thresh	1
MF-AGC			
Attack	2	**MIXER**	
Release	4	Low Band	(+)0.5
Make-Up Gain	4	Mid Band	(+)0.5
Gate Thresh	3	Pres Band	(-)4.0
RTP Speed	Slow	High Band	(-)5.0
RTP Level	(-)10		
		CLIPPER	
PR-AGC		Drive	(+)1.5
Attack	2		
Release	4	**COMP CLIP**	
Make-Up Gain	4	Drive	(+)1.5
Gate Thresh	2		
RTP Speed	Slow		
RTP Level	(-)10		
HF-AGC			
Attack	3		
Release	2		
Make-Up Gain	3		
Gate Thresh	1		
RTP Speed	Slow		
RTP Level	(-)5		